WE CLAIM:

SUB B5 a3

1. A system for receiving digital voice signals transmitted over a data network, comprising:

a jitter buffer, having a variable storage size, arranged to receive packets of data comprising the digital voice signals, to store said packets, and to serially output said packets;

a jitter buffer manager which (a) monitors the arrival said packets, (b) determines at least one variation parameter which measures the variation in transit delay among said arriving packets, and (c) controls the jitter buffer size in response to the variation parameter; and

a speed control module, which responds to a control signal from said jitter buffer manager by modifying a rate of consumption of packets serially output from said jitter buffer, to compensate for changes in said jitter buffer's storage size.

2. The system of claim 1, wherein variation parameter comprises a measure of variance between (a) actual packet arrival times, for a predetermined number of packets, and (b) a synchronous, average packet arrival time;

and wherein said jitter buffer manager controls said jitter buffer storage size in relation to said measure of variance.

3. The system of claim 2 wherein said buffer size is controlled to a size determined to allow capture of a predetermined fraction of packets within a predetermined time window, based upon the calculated measure of variance; and

wherein said predetermined fraction is less than or substantially equal to 1.

Subs a3

~~4. The system of claim 2 wherein said speed control module responds to a control signal from said jitterbuffer manager by reducing the rate of data consumption when said jitterbuffer size is increasing, while augmenting the data to maintain a predetermined rate of audio output.~~

5. The system of claim 4, wherein said data is augmented by selectively duplicating data corresponding to silent periods.

Subs a3

~~6. The system of claim 2, wherein said speed control module responds to a control signal from said jitter buffer manager by increasing the rate of data consumption when said jitterbuffer size is increasing, while selectively discarding data to maintain a predetermined rate of audio output.~~

7. The system of claim 6, wherein said jitter buffer manager selectively discards data corresponding to silent periods.

8. The system of claim 2 wherein said speed control module adjusts the rate of data consumption from said jitter buffer while maintaining audio output which substantially corresponds to natural human speech characteristics

9. The system of claim 1, further comprising:

an audio decoder, arranged to receive packets from said speed control module, to convert said packets into audio output.

Subs a3

~~10. A method of receiving digitally encoded, packetized audio transmitted across a data network, comprising the steps of:~~

~~monitoring the arrival times of said packets as they are received from the network;~~

~~loading said packets into a buffer having an adjustable size;~~

~~calculating an average packet delay relative to a synchronous serial output from said buffer;~~

~~calculating a time-varying variance parameter which quantifies deviations in packet delay from said average packet delay; and~~

~~adjusting said size of said buffer in response to a calculated value of said time-varying variance parameter.~~

11. The method of claim 10, comprising the further step of transferring said packets serially from said buffer at a variable rate to compensate for changes in size of said buffer.

12. The method of claim 11, comprising the further step of selectively modifying a decoded speech signal with a speed control module, to mask changes in said variable rate of transfer of said packets.

13. The method of claim 11, wherein said variance parameter is calculated as a sum of absolute values of deviations from a moving average of packet delay.

14. The method of claim 11, wherein said variance parameter is compared to a growth threshold, and said step of adjusting the size of said buffer comprises increasing said size when said variance parameter exceeds said growth threshold.

15. The method of claim 11, wherein said variance parameter is compared to a shrink threshold, and said step of adjusting the size of said buffer comprises decreasing said size when said variance parameter is less than said shrink threshold.

16. The method of claim 11, wherein said buffer size is adjusted to a size which will be statistically likely to accept a predetermined fraction of packets, based upon said calculated variance parameter, where said predetermined fraction is selected to produce a predetermined subjective audio quality in an audio signal decoded from said packets.

Subs a3

~~17. The method of claim 11, further comprising the step of:~~

~~comparing an average packet delay with a reference delay which corresponds to a temporally centered position in said buffer; and~~

~~adjusting said variable rate of transfer of packets from said buffer when said average packet delay deviates from said centered position by more than a threshold amount, thereby moving said centered position to align with said average packet delay.~~

add a3